IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of	:	Docket No.	0074-26485
POLETTI	:	Art Unit	2644
Application No. 09/673,808	:	Examiner	Corey P.Chau

Filed: January 12, 2001

For: An In-line Early Reflection Enhancement System for Enhancing Acoustics

DECLARATION

I, MARK ALISTAIR POLETTI, declare:

1. I am the MARK ALISTAIR POLETTI named as inventor in US patent application 09/673,808 titled "An In-line Early Reflection Enhancement System for Enhancing Acoustics".

2. I am employed as a research scientist at Industrial Research Limited in New Zealand which is a New Zealand Government owned research organisation. I hold a PhD (Acoustics) from the University of Auckland, New Zealand, 2001, and a MSc from the University of Auckland 1984, and BSc University of Auckland, 1982.

3. My areas of expertise include signal processing theory, digital signal processing, sound recording and reproduction systems, audio and acoustics systems. Attached marked MAP1 is a list of papers I have had published in peer reviewed journals and conference papers I have presented and I have also had two book chapters published.

4. In 2000 I received the New Zealand Royal Society award for the development of the VRA room acoustic enhancement system, which is licensed to the US company Level Control Systems (www.LCSaudio.com). In 1999 this system also won the Sound Product of the Year award, Lighting Dimensions International, Florida 1999.

5. I have carefully read US patent 5,812,674 to *Jot* et al (herein *Jot*). This discloses a system for simulating the acoustic quality of a virtual sound source in a virtual room. The result is a digital reverberator that simulates a virtual environment whose outputs are panned to control the localisation of sound. A large part of the patent discusses the generation of various parameters used in the reverberator and so is not relevant to the present application.

6. The present invention defined in the present application relates to a sound system for providing in-line early reflection enhancement. The system and method include an early reflection generation stage which generates a number of delayed discrete reproductions of the input signals. These delayed discrete reproductions of the signals are broadcast by loudspeakers into a room or other space.

   The early reflection generation stage includes a cross-coupling matrix and the early reflection generation stage has a unitary transfer function matrix such that the early reflection system has an overall power gain that is substantially constant with frequency. This improves stability when the early reflection system is used as part of a sound system.

7. *Jot* does not describe a processor for sound system that has an early reflection stage having a unitary transfer function matrix such that the processor has an overall power gain that is substantially constant with frequency to provide stability in the sound system.

8. Referring to Figure 9 of *Jot*, the room module system of the *Jot* system has two inputs ("Face","Omni") and seven outputs ("C", "L", "R", "S1", "S2", "S3", "S4"). The room module system has an overall transfer function matrix relating to the transfer function matrix between the two inputs and the seven outputs.

9. The overall transfer function matrix of the room module system shown in Figure 9 of *Jot* is not unitary. It does not have an overall power gain that is substantially constant with frequency.

10. Referring to Figure 9 of *Jot*, the room module system has a sub-system path between the Omni input and S1-S4 outputs. This sub-system path has a sub-system transfer function matrix.

11. The examiner has asserted that the *Jot* patent discloses an early reflection stage similar to that of the present invention, wherein the examiner asserts that the early reflection stage of *Jot* comprises the combination of a delay 731, a unitary mixing matrix 741, a delay 742 and a unitary mixing matrix 750. These components are shown in Figure 9 of *Jot*. These components form part of the sub-system path between Omni and S1-S4.

12. While the early reflection sub-system in *Jot* has similar components to those in the present invention, they are not arranged in such a way as to produce an overall transfer function matrix for the sub-system which is unitary, which is the purpose of the present invention.

13. Further, it is not meaningful to isolate the components of the *Jot* system noted in 10 and 11. These do not exist in isolation in a functional sense. These components form only part of the sub-system path between the Omni input and the four outputs S1 to S4. The sub-system path from Omni to S1-S4 also comprises equaliser 743, unitary mixing matrix 744, and delay 745 as shown in Figure 9

of *Jot.* All these components must be considered when evaluating the transfer function matrix of sub-system path Omni to S1-S4.

14. The transfer function matrix of the sub-system comprising the delay 731, the unitary mixing matrix 741, the delays 742, the downmixing of 8 to 4 signals at the outputs of delays 742, the equaliser 743 and the unitary mixing matrix 750 as shown in Figure 9 of *Jot* is not unitary . Further, the sub-system containing these elements plus the unitary mixing matrix 744, the delay lines 745, the regeneration of these delay line outputs back to 744, the downmixing of 8 to 4 signals at the outputs of delays 745 and the gain elements 746 added to the outputs of 743 as shown in Figure 9 of *Jot* is also not unitary.

15. I will demonstrate below that the overall transfer function matrix of the room module system is not unitary. I will also demonstrate that the transfer function matrix of the sub-system path from Omni to S1-S4 is not unitary. This will be demonstrated with reference to Figure 9 from *Jot,* which is redrawn below for clarity.

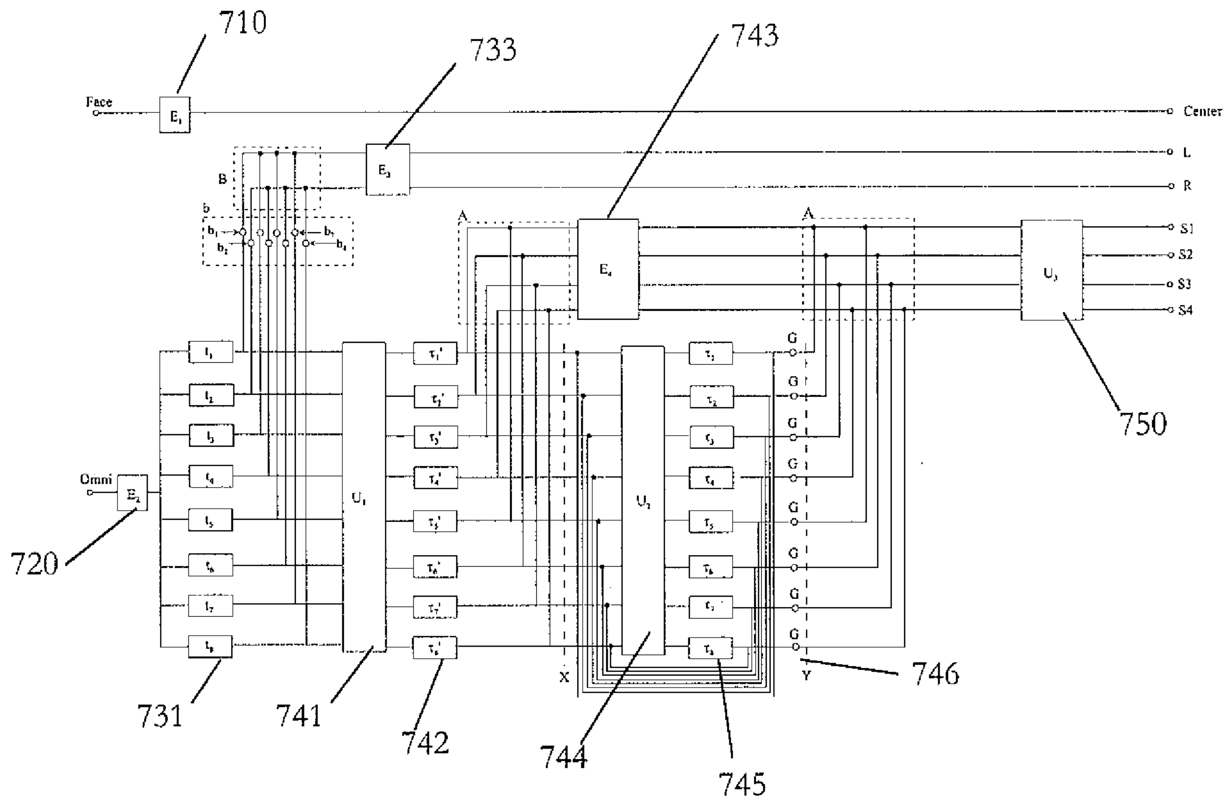

Figure 1: Redrawn version of Figure 9 from *Jot*

16. Figure 9 of US patent 5,812,674 shows a linear, digital processor which has two inputs (Face and Omni) and which generates seven outputs ("C", "L", "R", "S1", "S2", "S3", "S4"). The input delays t_1 to t_N are shown separately for clarity.

17. The properties of the room module are completely specified by the transfer function matrix **H**(z) which at each frequency describes the magnitude and phase of the transfer function from the *ith* input (*i=1,2*) to the *nth* output (*n = 1,2,3,4,5,6,7*). The transfer function matrix **H**(z) is the Z transform of the matrix of impulse responses from the from the *ith* input to the *nth* output. The Z transform is the standard method for analysing sampled digital systems and details on the Z transform may be found in [1]. Using the Z transform I will derive the various components of the room module, and then derive the overall transfer function matrix **H**(z). If **H** is unitary, then at each frequency $z = \exp\left[j2\pi f / f_s\right]$, $\mathbf{H}^H\mathbf{H} = \mathbf{I}$, where the superscript H denotes the conjugate

transpose of the matrix. If **H** is unitary at all frequencies then it has a power gain of one at all frequencies.

Analysis of room module components

Addition matrices

18. The addition of signals from the first set of delay lines t_1 to t_N after gains b_1 to b_N may be represented by the matrix

$$\mathbf{B} = \begin{bmatrix} 1 & 0 & 1 & 0 & 1 & 0 & 1 & 0 \\ 0 & 1 & 0 & 1 & 0 & 1 & 0 & 1 \end{bmatrix} \qquad 1$$

This adds the odd numbered channels to produce the L output and the even numbered channels to produce the R output, via equalisers $\mathbf{E}_3(z)$.

19. The addition of 8 signals from the second set of delay lines τ'_1 to τ'_N to four signals before the equalisers 743 may be represented by the matrix

$$\mathbf{A} = \begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 1 & 0 & 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 \end{bmatrix} \qquad 2$$

This same matrix also downmixes the outputs from the third set of delay lines τ_1 to τ_N.

First delay line

20. The first set of delay lines produces 8 delayed outputs from a single omni input. Each delay time is related to the number of delay taps and the sample rate, f_s, by

$$t_n = \frac{L_{tn}}{f_s} \qquad (3)$$

where L_{tn} is the number of taps from the omni input to the *nth* delay output.

The Z transform associated with a delay of L taps is z^{-L}. Hence, the transfer function matrix of the input delays may be represented as

$$\mathbf{H}_t(z) = \begin{bmatrix} z^{-L_{t1}} \\ z^{-L_{t2}} \\ z^{-L_{t3}} \\ z^{-L_{t4}} \\ z^{-L_{t5}} \\ z^{-L_{t6}} \\ z^{-L_{t7}} \\ z^{-L_{t8}} \end{bmatrix} \qquad (4)$$

In the simulations below we generate a monotonically increasing set of delays so that $L_{tn} > L_{t(n-1)}$.

21. The outputs of the delay line are fed via weighting gains b_1 through b_N to the L and R outputs. The weightings are not specified in the patent. However, since the amplitude of the sound pressure from a sound source reduces with the reciprocal of the distance, a subjectively valid form for these weightings is:

$$b_n = \frac{L_{t1}}{L_{tn}} \qquad (5)$$

so that longer delays correspond to larger attenuations, as would occur in a real room. The largest weighting value is b_1=1, and subsequent weightings are less than one.

<u>Second delay line</u>

22. The second set of delay lines receives 8 inputs and delays each to produce an output used to create secondary reflections (R_2). The transfer function matrix is therefore 8 by 8, and there is no cross coupling. The delays will be denoted $L_{r'n}$ and the transfer function matrix is:

$$\mathbf{H}_{r'}(z)=\begin{bmatrix} z^{-L_{r'1}} & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & z^{-L_{r'2}} & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & z^{-L_{r'3}} & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & z^{-L_{r'4}} & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & z^{-L_{r'5}} & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & z^{-L_{r'6}} & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & z^{-L_{r'7}} & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & z^{-L_{r'8}} \end{bmatrix} \qquad 6$$

At each frequency f, $z=\exp\left[-j2\pi f/f_s\right]=\exp\left(j\theta\right)$, and this matrix is unitary, ie

$$\mathbf{H}_{r'}{}^{H}\mathbf{H}_{r'}=\begin{bmatrix} \left|e^{-jL_{r'1}\theta}\right|^2 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & \left|e^{-jL_{r'2}\theta}\right|^2 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & \left|e^{-jL_{r'3}\theta}\right|^2 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & \left|e^{-jL_{r'4}\theta}\right|^2 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & \left|e^{-jL_{r'5}\theta}\right|^2 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & \left|e^{-jL_{r'6}\theta}\right|^2 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & \left|e^{-jL_{r'7}\theta}\right|^2 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & \left|e^{-jL_{r'8}\theta}\right|^2 \end{bmatrix}=\mathbf{I}$$

<u>Third delay line</u>

23. This has a similar form to the second delay line, with delays L_{tn}. The transfer function matrix is:

$$\mathbf{H}_\tau(z)=\begin{bmatrix} z^{-L_{\tau1}} & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & z^{-L_{\tau2}} & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & z^{-L_{\tau3}} & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & z^{-L_{\tau4}} & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & z^{-L_{\tau5}} & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & z^{-L_{\tau6}} & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & z^{-L_{\tau7}} & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & z^{-L_{\tau8}} \end{bmatrix} \qquad 7$$

Each delay line of length $L_{\tau n}$ must include a gain g_n less than one to ensure that the reverberant response (R3) decays away in time as in a natural room. The patent confirms this by referring to these delays as "absorbent delay banks." As shown in [2], a natural reverberant response is produced by ensuring that all poles have the same magnitude. This is achieved by setting:

$$g_n = e^{\frac{-3L_{\tau n}}{f_s RT}} = \gamma^{L_{\tau n}} \qquad 8$$

where RT is the reverberation time. We write these "loop gains" as a separate matrix

$$
\mathbf{g}=\begin{bmatrix}
\gamma^{L_{r1}} & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & \gamma^{L_{r2}} & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & \gamma^{L_{r3}} & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & \gamma^{L_{r4}} & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & \gamma^{L_{r5}} & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & \gamma^{L_{r6}} & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & \gamma^{L_{r7}} & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & \gamma^{L_{r8}}
\end{bmatrix}
$$

9

The complete transfer function matrix of the absorbent delay banks is thus $\mathbf{gH}_{\tau}(z)$.

<u>Unitary cross coupling matrices</u>

24. The room module has two 8 by 8 unitary mixing matrices and a 4 by 4 unitary output matrix for outputs S1 to S4. The exact values of these matrices are not specified. However, a matrix transfer function containing internal unitary matrices, and which is itself unitary, maintains this unitary property for all possible internal unitary matrices. Hence, the exact form of each unitary mixing matrix is non-critical. We therefore generate arbitrary mixing matrices. The first is the unitary circulant matrix

$$
\mathbf{U}_1=\begin{matrix}
0.3536 & 0.2500 & 0.5000 & 0.2500 & -0.3536 & -0.2500 & 0.5000 & -0.2500 \\
-0.2500 & 0.3536 & 0.2500 & 0.5000 & 0.2500 & -0.3536 & -0.2500 & 0.5000 \\
0.5000 & -0.2500 & 0.3536 & 0.2500 & 0.5000 & 0.2500 & -0.3536 & -0.2500 \\
-0.2500 & 0.5000 & -0.2500 & 0.3536 & 0.2500 & 0.5000 & 0.2500 & -0.3536 \\
-0.3536 & -0.2500 & 0.5000 & -0.2500 & 0.3536 & 0.2500 & 0.5000 & 0.2500 \\
0.2500 & -0.3536 & -0.2500 & 0.5000 & -0.2500 & 0.3536 & 0.2500 & 0.5000 \\
0.5000 & 0.2500 & -0.3536 & -0.2500 & 0.5000 & -0.2500 & 0.3536 & 0.2500 \\
0.2500 & 0.5000 & 0.2500 & -0.3536 & -0.2500 & 0.5000 & -0.2500 & 0.3536
\end{matrix}
$$

The second is the normalised Hadamard matrix

$$
\begin{matrix}
0.3536 & 0.3536 & 0.3536 & 0.3536 & 0.3536 & 0.3536 & 0.3536 & 0.3536 \\
0.3536 & -0.3536 & 0.3536 & -0.3536 & 0.3536 & -0.3536 & 0.3536 & -0.3536 \\
0.3536 & 0.3536 & -0.3536 & -0.3536 & 0.3536 & 0.3536 & -0.3536 & -0.3536
\end{matrix}
$$

$$\mathbf{U}_2 = \begin{array}{cccccccc} 0.3536 & -0.3536 & -0.3536 & 0.3536 & 0.3536 & -0.3536 & -0.3536 & 0.3536 \\ 0.3536 & 0.3536 & 0.3536 & 0.3536 & -0.3536 & -0.3536 & -0.3536 & -0.3536 \\ 0.3536 & -0.3536 & 0.3536 & -0.3536 & -0.3536 & 0.3536 & -0.3536 & 0.3536 \\ 0.3536 & 0.3536 & -0.3536 & -0.3536 & -0.3536 & -0.3536 & 0.3536 & 0.3536 \\ 0.3536 & -0.3536 & -0.3536 & 0.3536 & -0.3536 & 0.3536 & 0.3536 & -0.3536 \end{array}$$

The third is the unitary circulant matrix

$$\mathbf{U}_3 = \begin{array}{cccc} 0.2500 & 0.2500 & 0.2500 & -0.2500 \\ -0.2500 & 0.2500 & 0.2500 & 0.2500 \\ 0.2500 & -0.2500 & 0.2500 & 0.2500 \\ 0.2500 & 0.2500 & -0.2500 & 0.2500 \end{array}$$

We note that the above matrices have closely similar element magnitudes, which provides maximum cross coupling and hence diffusion when used in early reflection and reverberation processors [2,3].

Equalisation matrices

25. The Face and Omni inputs have single channel equalisers $\mathbf{E}_1(z)$ and $\mathbf{E}_2(z)$.

The mixed outputs from the first delay lines, and from the second delay lines, are equalised (733 and 743). These equalisers are 4 by 4 matrices:

$$E_3(z) = \begin{bmatrix} E_{31}(z) & 0 & 0 & 0 \\ 0 & E_{32}(z) & 0 & 0 \\ 0 & 0 & E_{33}(z) & 0 \\ 0 & 0 & 0 & E_{34}(z) \end{bmatrix} \tag{10}$$

and

$$E_4(z) = \begin{bmatrix} E_{41}(z) & 0 & 0 & 0 \\ 0 & E_{42}(z) & 0 & 0 \\ 0 & 0 & E_{43}(z) & 0 \\ 0 & 0 & 0 & E_{44}(z) \end{bmatrix} \qquad 11$$

Derivation of overall transfer function matrix

26. The overall transfer function matrix **H**(z) is a 7 by 2 matrix. The transfer function from Face input to C output is:

$$\mathbf{H}_{1,1}(z) = E_1(z) \qquad 12$$

27. The transfer function from the Omni input to the L and R outputs is the 2 by 1 matrix:

$$\mathbf{H}_{LR}(z) = \mathbf{E}_3(z)\mathbf{BbH}_t(z)\mathbf{E}_2(z) \qquad 13$$

with components $\mathbf{H}_{LR,1}(z)$ and $\mathbf{H}_{LR,2}(z)$.

28. The transfer function from the Omni input to the surround outputs S1 to S4 is the sum of the outputs from the second delay line (the secondary reflections) and the third delay line (the late reverberation). The transfer function matrix of the regenerative reverberator (from point X to point Y in figure 1) is:

$$\mathbf{H}_R(z) = \mathbf{GgH}_r(z)\left[\mathbf{I} - \mathbf{U}_2\mathbf{gH}_r(z)\right]^{-1}\mathbf{U}_2 \qquad 14$$

where **G** is an 8 by 8 diagonal matrix with the same gain in each channel:

$$
\mathbf{G} = \begin{bmatrix}
G & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & G & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & G & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & G & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & G & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & G & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & G & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & G
\end{bmatrix} \qquad 15
$$

and **I** is the 8 by 8 identity matrix..

29. The transfer function matrix from Omni to S1 through S4 is thus the 4 by 1 matrix:

$$
\begin{aligned}
\mathbf{H}_S(z) = {} & \mathbf{U}_3\mathbf{E}_4(z)\mathbf{A}\mathbf{H}_{\tau'}(z)\mathbf{U}_1\mathbf{H}_l\mathbf{E}_2(z) \\
& + \mathbf{U}_3\mathbf{A}\mathbf{H}_R(z)\mathbf{H}_{\tau'}(z)\mathbf{U}_1\mathbf{H}_l\mathbf{E}_2(z)
\end{aligned} \qquad 16
$$

with components $\mathbf{H}_{S,1}(z)$ through $\mathbf{H}_{S,4}(z)$.

30. The overall transfer function matrix $\mathbf{H}(z)$ has the form:

$$
\mathbf{H}(z) = \begin{bmatrix}
\mathbf{E}_1(z) & 0 \\
0 & \mathbf{H}_{LR,1}(z) \\
0 & \mathbf{H}_{LR,2}(z) \\
0 & \mathbf{H}_{S,1}(z) \\
0 & \mathbf{H}_{S,2}(z) \\
0 & \mathbf{H}_{S,3}(z) \\
0 & \mathbf{H}_{S,4}(z)
\end{bmatrix} \qquad 17
$$

31. If the room module is to have a constant power gain with frequency, then for $z = \exp\left[j2\pi f / f_s\right]$, $\mathbf{H}$(z) must be unitary: $\mathbf{H}^H\mathbf{H} = \mathbf{I}$. More generally, $\mathbf{H}$ may be unitary with a scale factor β, $\mathbf{H}^H\mathbf{H} = \beta\mathbf{I}$ for any real β. From **equation 17**:

$$\mathbf{H}^H\mathbf{H} = \begin{bmatrix} |E_1|^2 & 0 \\ 0 & |H_{LR,1}|^2 + |H_{LR,2}|^2 + |H_{S,1}|^2 + |H_{S,2}|^2 + |H_{S,3}|^2 + |H_{S,4}|^2 \end{bmatrix} \qquad 18$$

Hence, for $\mathbf{H}$ to be unitary, the input equaliser $\mathbf{E}_1$(z) must have a flat response with gain one (more generally β), and the sum of the squared magnitudes of the other six transfer functions must add up to one (more generally β).

Determination of power gain

32. The power gain of any system can be determined by assuming that the inputs are complex sinewaves with arbitrary magnitudes and phases. The discrete-time Face and Omni input signals may be represented as:

$$\begin{bmatrix} F(n) \\ O(n) \end{bmatrix} = e^{j2\pi fn/f_s} \begin{bmatrix} a_1 e^{j\phi_1} \\ a_2 e^{j\phi_2} \end{bmatrix} = e^{j2\pi fn/f_s}\mathbf{w} \qquad 19$$

where **w** is the 2 by 1 vector of complex input amplitudes and *n* is the discrete time index. The input power is the sum of the squared magnitudes of the two input signals:

$$P_{in} = a_1^2 + a_2^2 = \mathbf{w}^H\mathbf{w} \qquad 20$$

Note that by using complex input signals, the power is a constant value in time. An equivalent result is obtained by using real signals and averaging the squared outputs. The complex approach is a convenient way of obtaining the same result.

At the frequency f, the transfer function matrix $\mathbf{H}(z)$ has a value obtained by substituting $z = \exp\left[-j2\pi f / f_s\right] = \exp\left(j\theta\right)$ in **equation 17.**

33. The vector of 7 output signals is:

$$\begin{bmatrix} C\left(e^{j\theta}\right) \\ L\left(e^{j\theta}\right) \\ R\left(e^{j\theta}\right) \\ S_1\left(e^{j\theta}\right) \\ S_2\left(e^{j\theta}\right) \\ S_3\left(e^{j\theta}\right) \\ S_4\left(e^{j\theta}\right) \end{bmatrix} e^{j2\pi fn/f_s} = \mathbf{q}e^{j2\pi fn/f_s} = \mathbf{H}\left(e^{-j2\pi f/f_s}\right)\mathbf{w}e^{j2\pi fn/f_s} \qquad 21$$

where $\mathbf{q}$ is the vector of complex output amplitudes. The output power is:

$$P_{out} = \mathbf{q}^H\mathbf{q} = \mathbf{w}^H\mathbf{H}^H\mathbf{H}\mathbf{w} \qquad 22$$

where, as before, the power is constant in time due to the use of complex input signals. Hence the power gain is:

$$\frac{P_{out}}{P_{in}} = \frac{\mathbf{w}^H \mathbf{H}^H \mathbf{H} \mathbf{w}}{\mathbf{w}^H \mathbf{w}} \qquad 23$$

34. If the overall transfer function matrix $\mathbf{H}$ is unitary, then $\mathbf{H}^H \mathbf{H} = \mathbf{I}$ and

$$\frac{P_{out}}{P_{in}} = \frac{\mathbf{w}^H \mathbf{H}^H \mathbf{H} \mathbf{w}}{\mathbf{w}^H \mathbf{w}} = \frac{\mathbf{w}^H \mathbf{I} \mathbf{w}}{\mathbf{w}^H \mathbf{w}} = \frac{\mathbf{w}^H \mathbf{w}}{\mathbf{w}^H \mathbf{w}} = 1 \qquad 24$$

Hence, if $\mathbf{H}$ is unitary, the power gain of the room module is one at all frequencies. If $\mathbf{H}$ is a scaled unitary matrix, $\mathbf{H}^H \mathbf{H} = \beta \mathbf{I}$ for any real β, that the power gain is β which is also constant with frequency.

35. I will use complex input amplitudes with equal powers (magnitudes of one). I do this to ensure that both the Face and Omni signals have equal contribution to the power gain. If the Omni amplitude was very small, for example, then since the Face to C transfer function is one (for flat equalisation) the power through channel one would be approximately constant and this would give a false impression of the variation in the power gain.

I have already shown above **(equation 18)** that the first equaliser must be flat if the room module is to be unitary (since it is a single, un-cross-coupled channel it must have a flat response). I will also assume that the other equalisers have flat responses with frequency with magnitude one. Since an equaliser introduces a slowly varying amplitude with frequency, a non-flat equaliser will tend to increase the variation of power gain with frequency. Hence, flat equalisers will tend to *maximise* the chance of the room module having a constant power gain with frequency. (Patent US 5,812,674 uses non-flat equalisers for subjective reasons and the aim is not to produce a flat power gain with frequency.) The power gain of the room module will vary rapidly with frequency,

due to the effects of the delay lines. Hence, it is not possible for the equalisers to compensate for the power gain variation of the rest of the room module components.

36. I use delays

$L_\tau = 19, 23, 31, 41, 47, 61, 73, 89$

$L\tau' = 83, 101, 127, 151, 191, 227, 277, 331,$

and

$L\tau = 601, 797, 1069, 1399, 1801, 2251, 2789, 3373$

I use M=20,000 frequency samples from 0 Hz to $f_s/2$ Hz, and a sample rate of f_s=44100 Hz, which produces frequency samples 1.1 Hz apart.

37. As a first example, the power gain of the C output, the L and R outputs, the S1 to S4 outputs, and the overall power gain, are shown in Figure 2, for a reverberation gain G of zero, ie there is no late reverberation.

The C output power gain is one at all frequencies, since the equaliser gain is one, and there is no interaction with the other channels. The L and R outputs produce a fairly rapid variation of power gain due to the delay values in L_τ. The S1 to S4 outputs show a more rapid variation in power gain since $L_{\tau'}$ has larger delay values.

Clearly, the third graph in Figure 2 shows the total power at the S1-S4 outputs is not constant with frequency, hence the Omni to S1-S4 transfer function matrix is not unitary.

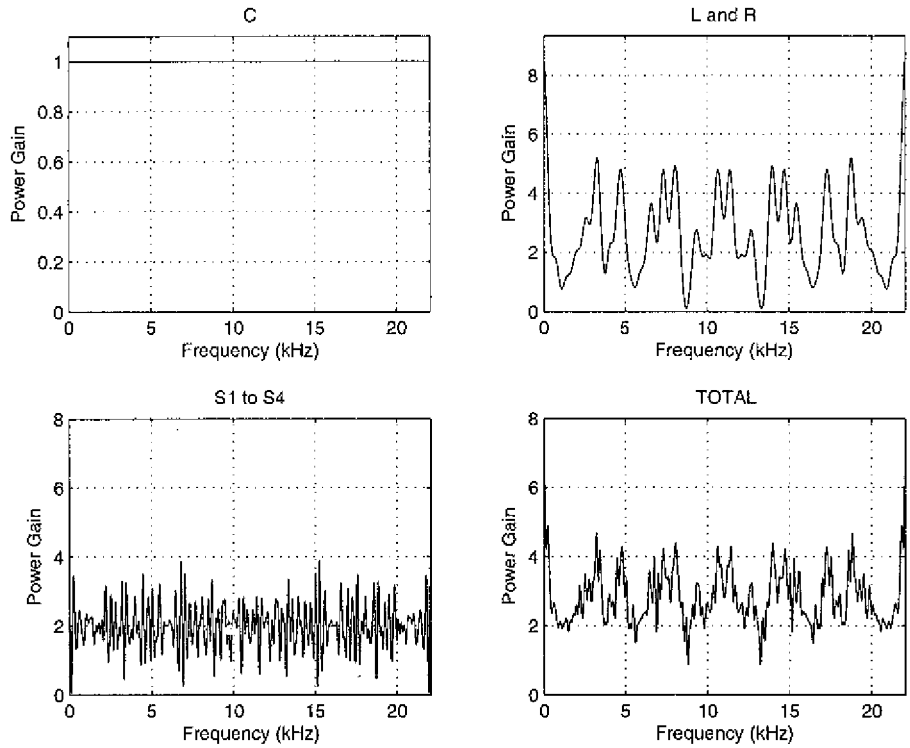

Figure 2: Power gains for G=0 (no late reverberation)

38. Figure 3 shows the power gains with the same parameters as above, but with a reverberation gain of G=0.15, and a reverberation time of 1 second. This increases the variation of the power gain at the S1 to S4 outputs due to the longer delay values in L_t, and the overall mean power gain is also increased.

The overall power gain is clearly not constant with frequency and therefore the room module is not useable in a sound system which includes both microphones and loudspeakers because it would increase the loop gain unacceptably at the peaks in its power gain.

Clearly, the third graph in Figure 3 shows the total power at the S1-S4 outputs is not constant with frequency, hence the Omni to S1-S4 transfer function matrix is not unitary.

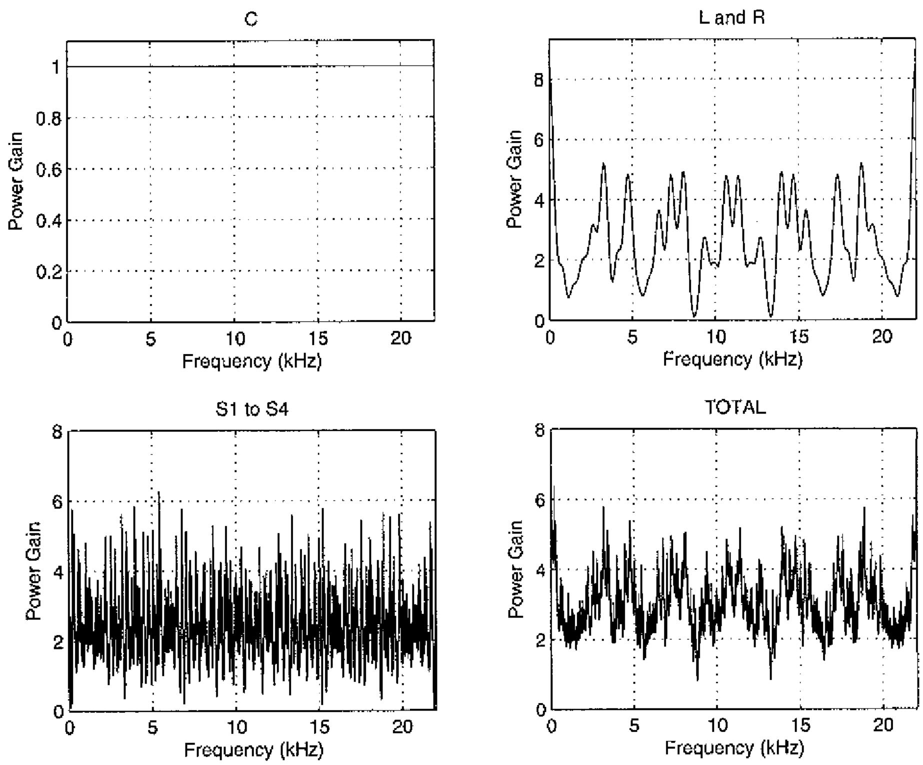

Figure 3: Power gains for G=0.15

39. Finally, Figure 4 shows the power gains for a different set of delays and unitary mixing matrices.

The delays are

L_t = 29, 31, 41, 47, 59, 71, 83, 101

$L\tau'$ = 101, 127, 149, 181, 223, 271, 331, 389

and

$L\tau$ = 701, 907, 1187, 1531, 1933, 2417, 2963, 3581

and the mixing matrices (obtained from QR decompositions of normal random matrices) are

-0.2450	0.3803	-0.0683	0.7847	-0.1191	0.1735	-0.0608	0.3563
0.0664	-0.5078	-0.1754	-0.0718	-0.0671	-0.2067	0.0842	0.8046

$$U1 = \begin{matrix} 0.1372 & -0.1645 & -0.6216 & -0.0429 & -0.2551 & 0.3660 & -0.5938 & -0.1196 \\ 0.0241 & 0.6020 & 0.0284 & -0.4781 & -0.5707 & -0.0750 & -0.0132 & 0.2760 \\ 0.3453 & -0.3137 & 0.4180 & 0.3277 & -0.6377 & -0.2049 & -0.1065 & -0.2008 \\ 0.1822 & 0.0748 & -0.6235 & 0.1688 & -0.1771 & -0.3218 & 0.5896 & -0.2478 \\ -0.4717 & 0.0157 & -0.1174 & 0.0233 & 0.0220 & -0.7462 & -0.4376 & -0.1188 \\ 0.7357 & 0.3222 & -0.0090 & 0.1108 & 0.3899 & -0.2920 & -0.2933 & 0.1388 \end{matrix}$$

$$U2 = \begin{matrix} -0.0505 & -0.0568 & -0.1470 & 0.1388 & -0.6167 & -0.3829 & 0.5308 & -0.3803 \\ 0.0661 & -0.7057 & -0.5514 & 0.0229 & -0.1848 & -0.0191 & -0.2171 & 0.3338 \\ 0.3048 & 0.1502 & -0.3461 & 0.7519 & 0.2386 & 0.3131 & 0.1889 & -0.0932 \\ -0.0206 & -0.4283 & -0.0876 & -0.2302 & 0.4688 & 0.0700 & 0.0632 & -0.7258 \\ 0.0935 & 0.4725 & -0.6161 & -0.2365 & 0.2157 & -0.4940 & -0.1963 & -0.0576 \\ 0.1930 & -0.2628 & 0.3579 & 0.3184 & 0.3715 & -0.6987 & 0.0652 & 0.1836 \\ -0.1553 & 0.0195 & 0.1241 & 0.3909 & -0.2623 & -0.1217 & -0.7546 & -0.3929 \\ -0.9108 & 0.0014 & -0.1543 & 0.2274 & 0.2356 & -0.0550 & 0.1389 & 0.1305 \end{matrix}$$

and

$$U3 = \begin{matrix} -0.7193 & -0.6914 & 0.0053 & 0.0672 \\ 0.0826 & -0.0635 & 0.9827 & 0.1530 \\ 0.0806 & -0.1772 & 0.1331 & -0.9718 \\ 0.6850 & -0.6975 & -0.1285 & 0.1664 \end{matrix}$$

40. We see from figure 4 that a different set of parameters produces a different variation in power gain to that in figures 2 and 3, but the power gain is still not flat. In addition, the power gain is increased by the larger variation in the unitary mixing matrices (which are not as optimal as the previous set of matrices).

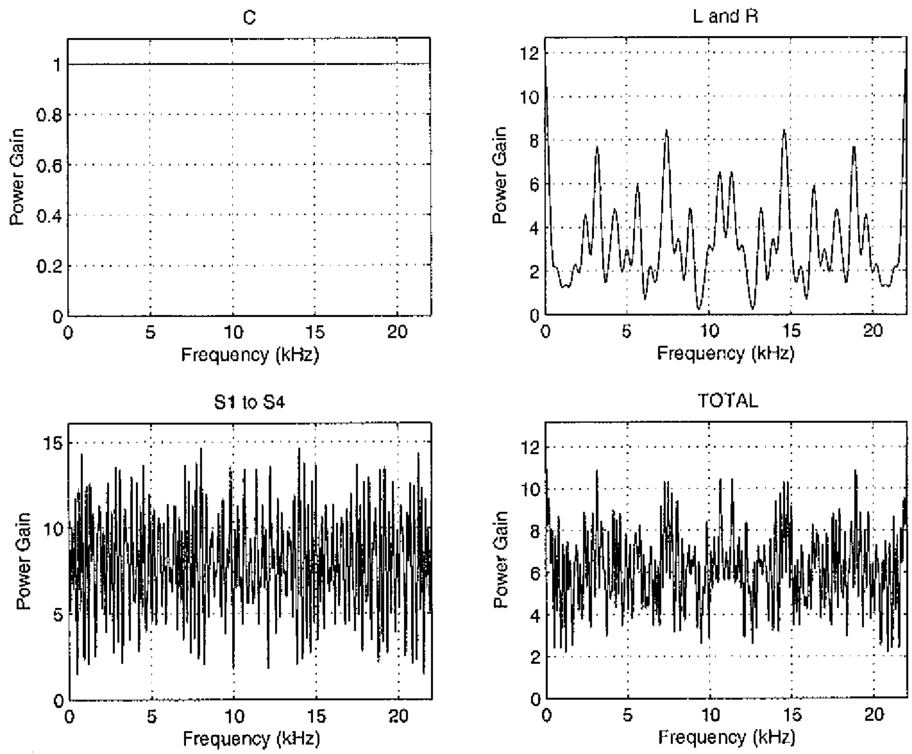

Figure 4: Power gains for G=.15 and a different set of delays and mixing matrices

Clearly, the third graph in Figure 4 shows the total power at the S1-S4 outputs is not constant with frequency, hence the Omni to S1-S4 transfer function matrix is not unitary.

41. The analysis above has shown that the room module in Figure 9 of US 5,812,674 does not have a flat power gain. This non-unitary behaviour will be observed for all possible delays and unitary mixing matrices, because the room module is inherently not a unitary structure.

The room module is not unitary because it adds multiple sets of signals from multiple sets of delay lines. The addition of signals in this manner will not produce a unitary overall transfer function matrix or constant power gain. As a simple example, consider a system with one input

and one output which is a sum of the input and a delayed version of the input, with L samples delay. The transfer function is

$$H(z) = 1 + z^{-L} \qquad 25$$

The magnitude squared frequency response for $z = \exp[j\theta] = \exp[j2\pi f / f_s]$ is

$$\left|H\left(e^{j\theta}\right)\right|^2 = 2\left[1 + \cos(L\theta)\right] \qquad 26$$

which is a comb filter response with a squared magnitude of 2 at frequencies $f = \left(k + \frac{1}{2}\right) f_s / L$ for integers k, and a squared magnitude of zero at frequencies $f = kf_s / L$. The response oscillates between 0 and 2 with frequency. There is no way to make the system unitary because the first component of the output (the direct signal) has magnitude one at all frequencies. *The key phenomenon in a unitary system is that when an output is the sum of several components, then if one component is large, the other components must be small, to maintain a constant power gain.* Channels two (L) to six (S4) of the room module are produced by summing components, and so the system is unable to produce this key phenomenon.

More complicated systems with more than one input and output and the summation of more signals simply produces more complicated forms of comb filter responses, as shown in figures 2 to 4.

The production of unitary overall transfer function matrices and flat power gains requires very specific forms of system, for example the series connection of unitary matrices with no summing of the outputs from each matrix, as in patent application 09/673808.

The room module in US 5,812,674 is not designed to produce a flat power gain, since it is not intended for use in a live sound system which includes both loudspeakers and microphones and

the risk of uncontrolled feedback. It appears to be well-suited for its intended use, but it does not obviate the Industrial Research patent application, which is designed specifically for live sound systems which include microphones. The Jot patent has components with unitary transfer function matrices, but the use of such matrices in reverberators is common, and predates the Jot patent (see for example [2]). The IRL patent application specifically discloses non-regenerative systems (no reverberation) which have unitary components, *and for which the overall transfer function matrix is also unitary.* Prior art processors do not have this overall-unitary property.

42. I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements are made with knowledge that wilful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such wilful false statements may jeopardize the validity of this Application for Patent or any patent issuing thereon.

DECLARED at Industrial Research Ltd
Lower Hutt, New Zealand)
this 17th day of January, 2007)

Mark Poletti

MARK ALISTAIR POLETTI

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